APPENDIX A
PROVISIONAL APPLICATION

SOUND CAPTURE DEVICE AND METHOD

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BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to devices and methods for capturing and processing acoustic sound waves.

2. Background

Virtually all microphone designs used today are passive designs that utilize technology developed in the 1960's. A conventional microphone puts out a low-level output signal on the order of milli-volts. This signal is transmitted to a processing circuit where it must be amplified to a line-level voltage (approximately 2-6 volts) before it can be used.

Conventional microphones are used in many applications including music, video and film recording systems, multi-media/entertainment applications, mobile/automotive applications, voice over internet, gaming devices, medical/transcription systems, and

computer/voice recognition systems. FIG. 1 shows a diagram of a typical professional studio example of a conventional microphone used in a typical recording system. This example of a typical professional recording system might be used in a recording studio for audio and/or film/video soundtrack purposes. In this example, a conventional condenser microphone 2 is connected to an operating power supply 4 and a microphone pre-amplifier 6. The conventional condenser microphone 2 is typically used in recording studios for such things as audio, video soundtrack, or broadcast. The function of the condenser microphone 2 is to convert acoustic sound waves into a corresponding electrical signal. Typically, in a the output from a conventional condenser microphone 2 is a low-level voltage of approximately .002 to .01 volts RMS. This low-level electrical signal that is output from the condenser microphone 2 is sent to the microphone pre-amplifier 6 via a microphone cable 32. Typically, in a professional studio recording system the microphone cable 32 can range from 20 to 100 feet in length. This length of cable is often required because of the distance between the audio source and the recording equipment, either in a studio or live concert setting.

While this example is of a typical professional studio recording system, the length of the cable connecting the microphone to the pre-amplifier can vary from a few inches to many feet or even be absent in cable-less systems.

In a conventional system, the purpose of the microphone pre-amplifier 6 is to increase the low-level electrical signal output by the condenser microphone 2 (approximately .002 to .01 volts RMS) to approximately 2 to 4 volts. The magnitude of this gain (approximately 40 to 60 dB) can be adjustable, depending upon the application. For example, a female singer with a soft voice might require a higher gain than would a drum set in a rock band. The microphone pre-amplifier 6 is usually either transformer-coupled or capacitor-coupled.

In the typical professional studio recording system using a conventional microphone, the power supply 4 is typically a 48 volt DC power supply. The purpose of the power supply 4 is to supply operating power for the condenser microphone 2 and the microphone pre-amplifier 6. In a typical application, the power supply is usually single-ended and is often called a "phantom power supply." In other applications, the

power supply can vary in operating specifications depending on the application.

The microphone pre-amplifier 6 is connected to the microphone gain 8. The purpose of the microphone gain 8 is to adjust the master volume of the audio signal. After the master volume is adjusted in the microphone gain 8, the electrical signal is amplified by a line pre-amplifier 10 to provide approximately 10 dB gain to the signal.

In a conventional professional studio recording system, the recording signal is next sent to an equalizer 12. The purpose of the equalizer 12 is to alter the frequency response in the recording signal as desired to provide specific sound quality.

In a conventional professional studio recording system, the equalizer 12 is often connected to a limiter 14. The purpose of a limiter 14 is to lower (i.e., to "compress" or "limit") the dynamic range of the recording signal. The limiter 14 is connected to an effects pre-amplifier 16. If desired, the effects pre-amplifier 16 can be used by the recording professional to provide specific sound effects in the recording signal, such as an "echo" or other "delay" effect.

In a typical professional studio recording system, the electrical recording signal is sent from the effects pre-amplifier 16 to a phase circuit 18. Often, a phase circuit 18 is required to adjust the signal phase because other processing causes the original signal to be out of phase. In this typical application, the phase circuit 18 is connected to a pan pot 20. Because most conventional condenser microphones 2 are monaural, the pan pot 20 circuit serves as a "balance control" such that when a mono recording signal is input, the pan pot 20 provides a left and right signal of varying degrees. The input mono recording signal can then be played as a stereo sound for the listener.

After proceeding through the pan pot 20, the electrical recording signal then passes to a summing pre-amplifier 22. In a typical multi-track channel recording system, the summing pre-amplifier 22 sums the left and right channels of the input tracks. The summing pre-amplifier 22 is connected to the master gain 23, which controls the volume of the mixed recording signal.

In the typical recording system, the signal is then passed through to another line pre-amplifier 24

where the signal can be amplified approximately 10 to 20 dB. Next, the signal is sent to an analog to digital converter (A/D converter) 26. In the A/D converter 26, the input analog recorded signal is quantized and is converted into a digital signal, usually comprising 16 to 24 bits. Next, the signal is sent to a digital recorder 28 for recording and use in various formats.

The conventional microphone 40 used in a typical professional studio recording system application has several inherent drawbacks. For example, the cable 42 that the electrical signal must travel along before amplification takes place allows for degradation and attenuation of the signal before the amplification. The cable 42 also introduces potential radio frequency and power supply interference.

Furthermore, conventional microphones 40 are often unable to process accurate phase information and frequency response due to poor off-axis performance. A conventional microphone will generally exhibit its best performance if it is used on-axis, i.e., oriented directly in front of the sound source. Off-axis refers to an audio or sound source that is not directly in front of a microphone transducer. When an off-axis

audio signal is sent to a conventional microphone, signal degradation and a change in the frequency response of the reproduced audio signal results.

FIG. 2 shows a diagram illustrating another example of a conventional microphone as used in a typical computer voice recognition application. In this example, a conventional microphone 40 is connected by a cable 42 to a microphone pre-amplifier 46, which is on a conventional sound card mounted inside a personal computer 44. Typically, the length of cable 42 is approximately six feet. However, the cable 42 can vary from a few inches to several feet in length, or is missing entirely in cable-less systems. The microphone pre-amplifier 46 is connected to an A/D converter 48, which is connected to a voice recognition circuit 50.

In operation, the conventional microphone 40 converts acoustic sound waves into a corresponding electrical signal. Typically, the output from this conventional microphone 40 is approximately .002-.01 volts RMS. This relatively low-level electrical signal must then pass through the cable 42 before being amplified by the microphone pre-amplifier 46. Subsequently, the signal is converted from analog to

digital in the A/D converter 48. Finally, the signal passes to the voice recognition circuit 50 for processing of the signal to identify the words being spoken.

One of the problems that exists in voice recognition systems is that for all conventional microphones 40 currently available, including so-called array microphones, suffer from poor off-axis performance and are unable to deliver accurate signals, especially in the presence of ambient noise. Further, the conventional microphone 40 converts analog voice signals to a low-level electrical signal that is typically required to travel over a varying distance (of a few inches to six feet or more) to a microphone preamplifier 46 located on a computer sound card. During its journey to the preamplifier, this low level signal is very susceptible to further degradation and interference (e.g., radio frequency interference and power supply distortion from the computer processor and monitor which degrade the signal).

The low-level signal is amplified by the computer's sound card, which typically contains a single-stage microphone pre-amplifier. Because this single-stage device significantly increases the signal

gain (40 to 60 dB) and limits the signal bandwidth, additional noise is introduced and the signal suffers reduced slew rate. This poor quality signal is then fed to the A/D converter, which sends the resultant poor quality digital signal to a voice recognition circuit 50.

Therefore, there is a need for a sound capture device and method that offers improved sound quality and accuracy in various applications.

SUMMARY OF THE INVENTION

The present invention provides a sound capture device comprising one or more transducers for converting an audio signal to a low-level electrical signal and directly providing this signal to an amplifier for amplifying to a line-level electrical signal. In one embodiment of the present invention, a power supply is directly connected to the amplifier for supplying operating power that is independent of the audio signal.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be described with particular embodiments thereof, and references will be made to the drawings in which:

FIG. 1 described above, shows an example of a conventional microphone as used in a typical professional studio recording system;

FIG. 2 described above, shows a conventional microphone as used in a typical personal computer voice recognition application;

FIG. 3 shows a sound capture device in an embodiment according to the present invention;

FIG. 4 shows a top view of a amplifier and a transducer device in a sound capture device in an embodiment according to the present invention;

FIG. 5 shows a sound capture device in an embodiment according to the present invention as used in a typical professional studio recording system application;

FIG. 6 shows a schematic representation of a transducer device and a amplifier in a sound capture device in an embodiment according to the present invention;

FIG. 7 shows a schematic drawing of a power supply in a sound capture device in an embodiment according to the present invention;

FIG. 8 shows a sound capture device in an embodiment according to the present invention as used in a typical voice recognition application; and

FIG. 9 shows a graphical representation of off-axis performance of the sound capture device in an embodiment according to the present invention.

DETAILED DESCRIPTION

FIG. 3 illustrates a sound capture device 100 in an embodiment according to the present invention. In this embodiment, a transducer 102 is connected to a amplifier 104. The purpose of the transducer 102 is to convert acoustic sound waves into an electrical signal. While this embodiment shows a single transducer 102, other embodiments can comprise a plurality of transducers to provide a plurality of electrical signals. The transducer 102 can comprise various embodiments, such as one or more condensers or dynamic or electret microphones. In addition, the transducer device 102 can provide mono, binaural, stereo, multiple channel or phased-array inputs.

In a typical example, the transducer device 102 provides approximately a .13 volt output at an input of 96 dB of volume. Various other examples of the present invention are possible such that the output level is a function of the input level. The purpose of the amplifier 104 is to convert this relatively low-level electrical signal into a much higher and more accurate line-level signal. Typically, the amplifier 104 will boost an input signal of approximately .13 volts to a line-level voltage that averages 2 to 6 volts RMS, but is capable of providing 0 to 9 volts. Other embodiments of the present invention can operate at higher voltages, depending upon the application. This line-level signal is output through line out 108 for interpretation and use of the signal by various types of systems using audio electronic signals.

One advantage of the present invention is related to the relatively close distance between the transducer device 102 and the amplifier 104. In a conventional microphone system, the microphone is typically separated from the pre-amplifier by a length of cable (shown in FIG. 1). This length of cable can be from a few inches to over 100 feet in length, which can create distortions that make the cable physical qualities

important (e.g., capacitance, inductance and wire insulation). These distortions are magnified when the relatively small magnitude of the electrical audio signal travels the length of this cable before amplification. Thus, in conventional systems, these distortions are amplified by the pre-amplifier along with the electrical audio signal.

In contrast, in the present invention, the need for a long portion of cable before the amplifier is eliminated by placing the amplifier 104 in a much closer physical proximity to the transducer device 102. This allows the initial signal to be immediately amplified to a line-level voltage. Accordingly, because noise introduced by a long cable is not amplified in the present invention, the signal that is amplified by amplifier 104 has much less distortion than that found in conventional microphone devices. Further, the present invention provides less distortion and interference than conventional systems using a cable-less microphone.

In an embodiment of the present invention, the power supply 106 provides electrical operating power through line 110 to the amplifier 104. Typically, the power supply 106 is DC and produces very low noise. In

conventional microphone systems, the power supply line is typically connected to the line out (shown in FIG. 1), thus causing distortion and interference with the electrical audio signal. In an embodiment of the present invention, however, the power supply line 110 is separate from the line out 108, thus eliminating this source of error and distortion. Importantly, the electronic audio signal is modulated independently of the power supply input.

FIG. 4 illustrates a top-view diagram of the transducer device 102 and the amplifier 104 of FIG. 3 in an embodiment according to the present invention. In this embodiment, the transducer device 102 is comprised of two condenser elements 112 connected by a connector 114.

In this embodiment, the two condenser elements 112 are fixed at a 15 degree angle outboard of a center line that is perpendicular to the connector 114. Further, in this embodiment, the condenser elements 112 are positioned at a 30 degree offset from each other. The 30 degree offset of the condenser elements 112 from each other seeks to imitate how human ears actually receive sound waves. This offset allows better off-axis performance than conventional microphones. While

30 degrees from each other is the degree of offset in this embodiment, other embodiments of the present inventions are possible including, but not limited to, 0 degrees to 90 degrees. In addition, the specific angle or angles that the condenser elements 112 are offset can be adjusted to provide various sound qualities.

In this embodiment, the condenser elements 112 are separated by 12 inches in distance (i.e., the length of connector 114). For the average person the linear distance through the skull between the two ears is approximately 7 to 8 inches. For sound waves traveling toward the head at any angle up to 180 degrees from straight ahead (the axis), since sound waves cannot appreciably travel through the head and therefore must travel around the head to the far ear, the effective distance between the two ears that the sound waves must travel is approximately 12 inches (i.e., based on the formula that distances equal pi multiplied by the radius). Accordingly, in this embodiment, the two condenser elements 112 are separated by 12 inches in distance. While 12 inches is the amount of separation in this embodiment, other embodiments of the present inventions are possible including, but not limited to,

zero separation of the condenser elements 112 up to many feet of separation (e.g., when placed in various positions around a concert stage). This enables the condenser elements 112 to capture (and ultimately reproduce upon playback) the phase angle information and the imaging at the same ratio as the person listening to a particular sound live. The result is a more realistic reproduction of recorded sound than is possible with conventional microphone devices.

In this embodiment, connector 114 can comprise various shapes and materials, but typically can be 1/4" wide or smaller so as not to block the incoming sound waves to the condenser elements 112. In other embodiments, connector 114 can be various sizes, or can be completely absent in wireless embodiments.

FIG. 5 shows a diagram of a sound capture device 100 as used in a typical professional studio recording system application. In this embodiment, acoustic sound waves are received by the condenser elements 112 which provides a low-level electronic audio signal to the amplifier 104. The amplifier 104 boosts the low-level signal to a line-level signal (e.g., 2 to 6 volts) and outputs this high-level signal on line out 108 to a volume control 116. Volume control 116 is a variable

potentiometer that typically can comprise components including, but not limited to, line gains or microphone gains. After the volume control 116 adjusts the volume of the recording signal, the signal passes to a summing amplifier 118 that sums the left and right channels of the input tracks to provide a master electrical signal. In this example of a recording system application, the master electrical signal then passes to a master volume control 120 for volume control. Next, the signal passes to a line amplifier 122 for further amplification, typically on the order of 10dB. Subsequently, the signal passes to an A/D converter 124 where it is converted from an analog signal into a digital signal. In this example, the A/D converter 124 comprises 24 bits, but can be many other types, including, but not limited to, 1, 16, 20 or 24 bits, or other number depending upon the specific converter technology in use. Finally, the recording signal passes to a digital recorder 126 for recording. The digital recorder can then provide the digital signal output to various devices for playbacks or other uses. In another embodiment of the present invention, an analog recorder (not shown) could be used in place of the A/D converter 124 and digital recorder 126 to

record the recording signal. It is understood that while FIG. 5 shows the sound capture device 100 used in a typical professional studio recording application, other embodiments are possible such that the order of the other components shown can be different, and/or components can be added or taken away from the system.

Compared to a conventional microphone in a typical professional studio recording application (shown in FIG. 1), the sound capture device 100 as used in a typical recording application produces a much higher quality signal for recording. For example, in one embodiment of the present invention, the 30 degree offset and 12 inch separation of the condenser elements 112 from each other allows a more realistic reproduction of phase information and better off-axis performance than conventional microphones. Also, the line out 108 provides a much higher output signal that is sent to the microphone gain 116 (e.g., 2 to 6 volts), as compared to the signal sent to the microphone gain 8 in FIG. 1 (e.g., .002 to .01 volts). Further, the power supply 106 in the sound capture device 100 provides a much lower noise level power supply than does the operating power supply 4 in FIG. 1.

In addition, this embodiment of the sound capture device 100 used in a recording application eliminates the need for various components of a typical recording system that uses a conventional microphone. For example, the line pre-amplifier 10, equalizer 12, limiter 14, effects pre-amplifier 16, phase amplifier 18 and pan pot 20 (shown in FIG. 1) can be eliminated by the embodiment of the sound capture device 100 shown in FIG. 5. These components can become unnecessary because a much higher quality electrical signal is provided by the sound capture device 100 to the volume control 116 in FIG. 5 than is provided by the conventional microphone 2 and microphone pre-amplifier 6 to the microphone gain 8. However, if desired, these components can be used alone or in combination with other embodiments of the present invention to provide higher sound quality provided by conventional systems.

FIG. 6 shows a schematic diagram of the condenser elements 112 and amplifier 106 of the sound capture device 100 in an embodiment of the present invention. Typically, in this embodiment, the condenser elements 112 comprise P9959-ND or WM-60 AY capsules. In this embodiment, the condenser element 112 is connected to resistor 128 and capacitors 132 and 134. Typically,

the value of resistor 128 is 4.99 kilo-ohms, capacitor 132 is a .4 micro-Farad/200 volt polypropylene capacitor and capacitor 134 is a .01 micro-Farad/50 volt polystyrene capacitor. Capacitors 132 and 134 are connected to resistor 138 and the positive input of amplifier 140. The negative input to amplifier 140 is connected to resistors 136 and 148 and capacitor 146. The output of amplifier 140 is connected to capacitor 146 and resistors 148 and 156. Typically, the value of resistor 138 is 20 kilo-ohms, the value of resistor 136 is 2 kilo-ohms, resistor 148 is 10 kilo-ohms, resistor 156 is 499 ohms and capacitor 146 is a 15 pico-Farad/50 volts polystyrene capacitor. Resistor 156 is connected to the positive input of amplifier 162 while the negative input is connected to resistors 152, 150 and 160 and capacitor 158. Typically, amplifiers 140 and 162 are of a high quality construction such as found in a OPA 627 AP amplifier made by Burr Brown. A switch 154 is connected between resistor 150 and resistor 152. The output of amplifier 162 is connected to resistor 160 and capacitor 158, as well as to resistor 168, which is connected to line out 108. In this example, typical values of the components include resistor 150 at 4.99 kilo-ohms, resistor 152 at 4.99 kilo-ohms,

resistor 160 at 10 kilo-ohms, resistor 168 at 249 ohms and capacitor 158 is a 15 pico-Farad/50 volt polystyrene capacitor. Power supply 130 converts a power supply received from power supply 104 (not shown in FIG. 6) to approximately 6 volts for operating power for the condenser elements 112. Power supplies 142, 144, 164 and 166 provide operating power for the amplifiers 140 and 162. In this example, power supplies 142 and 164 are +14 volts and power supplies 144 and 166 are -14 volts. In other embodiments, different operating power supplies can be provided, depending upon the application.

In operation, the condenser element 112 provide a signal to capacitors 132 and 134, which act as coupling capacitors to block any DC component created by power supply 130 and allow passage of the audio component of the signal sent by condenser element 112. After capacitors 132 and 134, the DC component of the signal is again filtered by resistor 138 before being input into amplifier 140. In this embodiment, amplifier 140 provides a first stage gain of approximately 15 dB to the signal, which is then outputted through resistor 156 to amplifier 162. Capacitor 146 and the voltage dividing network comprising resistors 136 and 148 are

adjustable as desired to change the feedback signal to amplifier 140.

As the second stage of amplification, amplifier 162 also provides approximately 15 dB of gain to the signal, which is outputted through resistor 168 on line out 108. Thus, this embodiment of the present invention provides a two-stage amplification of approximately 15 dB gain for each stage of the amplifier 104 for a total gain of approximately 30 dB. By using two stages of amplification rather than one, the system produces a more accurate final output signal. While this gain is typical, the gain can be adjusted to provide other levels of gain depending on the desired sound quality or application.

In this embodiment, the feedback loop for amplifier 162 comprises resistors 150, 152 and 160 and capacitor 158. Switch 154 is provided to allow an adjustable gain depending on whether the switch 154 is open or closed. If switch 154 is closed, resistors 150 and 152 act in parallel and provide approximately 14 dB of gain. This situation can be useful for low volume level recordings. If switch 154 is open, only resistor 152 remains in the circuit and in this example, approximately 9.5 dB of gain is provided, which can be

used when high volume level recording applications are desired. Finally, in this example, a line-level voltage of approximately 2 to 6 volts RMS is output on line out 108 for use in the professional studio recording system application shown in FIG. 5.

FIG. 7 illustrates a power supply 106 in an embodiment of the present invention. Resistor 168 is an output buffer for current limiting protection in a situation of faulty cable that is shorted. In this embodiment fuse 170 is connected to switch 172, which is connected to the primary winding of transformer 174. In this example, the transformer 174 is a 25 volt AC center-tap transformer. Also, a typical value of fuse 170 is a 250 micro-amp/250 volt fuse, and switch 172 is a 3 amp/250 volt switch. The secondary winding of transformer 174 is connected to a 4-way bridge rectifier 176, which can typically comprise four 1N 4944 diodes. The rectifier 176 is connected to resistors 186 and 188 and capacitors 178 and 182. Resistors 186 and 188 are connected to capacitors 180 and 184, respectively, and to resistors 194 and 196. Typically, values for capacitors 178, 180, 182 and 184 are 3,300 micro-Farad/35 volt capacitors, and resistors 186 and 188 are 10 ohms/2 watt resistors. Resistor 194

is connected to zener diode 190, capacitors 202 and 204 and transistor 210. Resistor 196 is connected to zener diode 192, capacitors 198 and 200 and transistor 212. Resistor 206 is connected between capacitor 204 and transistor 210. A typical value for capacitors 198 and 202 is 2200 micro-Farad 25 volt capacitor, and for capacitors 200 and 204 is a .1 micro-Farad/200 volt capacitor. A typical diode for zener diodes 190 and 192 is 1N 5245B, which is a 15 volt diode, and for the resistors 194 and 196 is 4.99 kilo-ohms/.25 W. Resistor 208 is connected between capacitor 200 and transistor 212. Transistor 210 is also connected to transistor 218 and resistor 216, while transistor 212 is connected to transistor 220 and resistor 214. In this example, transistor 210 can be a KN 4401, transistor 212 is a KN 4403, transistor 218 is a MJE 182 and transistor 220 is a MJE 172. Typically, resistors 206 and 208 are 100 ohms while resistors 214 and 216 are 20 ohms. Capacitors 222 and 224 are connected to each other in series, as well as connected between transistor 218 and 220. Transistor 218 is connected to capacitors 226 and 228 and resistor 236. Transistor 220 is connected to capacitors 230 and 232 and resistor 234. Typically, capacitors 222 and 224

are .1 micro-Farad/200 volt capacitors and the value of resistors 234 and 236 is 10 kilo-ohms/.25 watts. A typical value for capacitors 226 and 230 is a 4 micro-Farad 200 volt capacitor, and for capacitors 228 and 232 is a .1 micro-Farad 200 volt capacitors.

In operation, a standard wall outlet power supply of 120 volts AC (not shown) is input to the power supply 106 at nodes 242 and 244. Transformer 174, switch 172 and the 4-way bridge rectifier 176 converts the 25 volts AC to 17.625 x 2 volts DC. In other embodiments, other input voltages such as 220 volts AC can be utilized along with a corresponding transformer 174. In this embodiment, the purpose of capacitors 178, 180, 182 and 184 and resistors 186 and 188 is to provide filtering for the DC signal. After being smoothed by these capacitors, the signal next passes through zener diodes 190 and 192 and resistors 194 and 196, which provide a 15 volt reference voltage. The signal then passes through capacitors 198, 200, 202 and 204 in order to eliminate noise from the voltage reference. Next, the signal passes to transistors 210, 212, 218 and 220 for final regulation of the voltage to approximately 13.8 volts in this example. Resistors 206, 208, 214 and 216 provide stabilization for the

circuit. In addition, capacitors 222 and 224 provide dampening to increase higher frequencies because of the drop off in high frequencies caused by the transistors. Capacitors 226, 228, 230 and 232 are also used for this dampening purpose. In this example, the final regulated power output at node 238 is +13.8 volts DC and node 240 is -13.8 volts DC. In other embodiments, the final regulated power output can vary depending on the specific application. Resistors 234 and 236 acts as bleed resistors such that the circuit is discharged to 0 volts if the standard wall outlet power supply (not shown) is unplugged.

Compared to a conventional microphone in a typical recording application (shown in FIG. 1), the sound capture device 100 used in a typical professional studio recording application produces a much higher quality signal for recording. For example, the 30 degree offset and 12 inch separation of the condenser elements 112 from each other allows a more realistic reproduction of phase information and better off-axis performance than conventional microphones. Also, the line out 108 provides a much higher output signal that is sent to the volume control 116 (e.g., 2 to 6 volts), as compared to the signal sent to the microphone gain 8

in FIG. 1 (e.g., .002 to .01 volts). Further, the power supply 106 in the sound capture device 100 provides a much lower noise level power supply than does the operating power supply 4 in FIG. 1.

In addition, this embodiment of the sound capture device 100 used in a professional studio recording system application eliminates the need for various components typically found in a professional studio recording system using a conventional microphone. For example, the need for a line pre-amplifier 10, equalizer 12, limiter 14, effects pre-amplifier 16, phase amplifier 18 and a pan pot 20 (shown in FIG. 1) can be eliminated (if desired) by the embodiment of the sound capture device 100 shown in FIG. 5. These components can be eliminated because of the much higher quality electrical signal provided by the sound capture device 100 to the volume control 116 in FIG. 5 than is provided by the conventional microphone 2 and microphone pre-amplifier 6 in FIG. 1.

FIG. 8 illustrates a diagram of a sound capture device 100 in a typical personal computer voice recognition application in another embodiment according to the present invention. In this embodiment, a single condenser element 246 is connected to a amplifier 104.

In other embodiments, condenser element 246 can comprise other transducer elements such as dynamic or electret or phased-array devices. The amplifier 104 is connected to power supply 106 and to A/D converter 250 through line out 248. A/D converter 250, which is on a conventional sound card inside of a personal computer 254, is connected to a voice recognition circuit 252.

In operation, the condenser element 246 converts acoustic sound waves into a corresponding electrical signal. Typically, the output from the condenser element 246 is approximately .002-.01 volts RMS. This low-level electrical signal is then immediately amplified by the amplifier 104 to a line-level voltage of approximately 2 to 6 volts.

In this embodiment in a voice recognition application, the line-level voltage signal is provided to the A/D converter 250 for conversion from an analog signal to a digital signal. Finally, the signal is sent to the voice recognition circuit 252 for processing of the signal to determine the spoken words. In other applications of the present invention, the recognition system can be used for music or other sounds.

One of the advantages provided by the embodiment of the sound capture device 100 shown in FIG. 8 is that it offers superior off-axis performance as compared to conventional microphones. Typically, as discussed in the background, off-axis performance of a conventional microphone is very poor. As shown in FIG. 9, in one embodiment of the transducer device 112, the off-axis performance of the condenser element 246 is demonstrated to be a maximum of -2 dB from 125 Hz to 8 kHz and a maximum of -4 dB from 125 Hz to 18 kHz when measured up to 90 degrees off-axis, which is substantially better performance than found in conventional microphones.

Further, in this embodiment the sound capture device 100 eliminates noise in the signal by placing the amplifier 104 physically closer to the condenser element 246, typically only a few inches. In contrast, conventional microphone 40 must send a low-level electrical signal over a distance of up to several feet to a microphone pre-amplifier 46 located on a computer sound card (shown in FIG. 1). During this much longer path to the amplifier 46, this low level signal is susceptible to further degradation and interference, including radio frequency interference and power supply

distortion. Thus, while in the conventional microphone system, the low-level signal is amplified along with the accumulated noise, the sound capture device 100 of the present invention provides for amplification of the clean signal at a much earlier point. This poor quality signal is then fed to the A/D converter, which sends the resultant poor quality digital signal to a voice recognition circuit 50.

The present invention has been described with respect to particular embodiments thereof, and numerous modifications can be made which are within the scope of the invention as set forth in the claims.

What is claimed is:

1. A sound capture device, comprising:

one or more transducers for converting an audio signal to a low-level electrical signal and directly inputting the low-level electrical signal to an amplifier connected to the one or more transducers for amplifying the low-level electrical signal to a line-level electrical signal.

2. The device of claim 1, wherein the one or more transducers comprises one condenser element.

3. The device of claim 1, wherein the one or more transducers comprises two or more condenser elements.

4. The device of claim 3, wherein the two or more condenser elements are separated by a predetermined distance.

5. The device of claim 4, wherein the predetermined distance is approximately 12 inches.

6. The device of claim 1, wherein the amplifier provides two stages of amplification.

7. The device of claim 6, wherein each of the two stages of amplification provides a gain of approximately 15 dB.

8. The device of claim 3, wherein the two or more condenser elements are offset from each other at an angle of approximately 0 degrees to 90 degrees.

9. The device of claim 8, wherein the two or more condenser elements are offset from each other at an angle of approximately 30 degrees.

10. A sound capture device, comprising:

one or more transducers for converting an audio signal to a low-level electrical signal and directly inputting the low-level electrical signal to an amplifier connected to the one or more transducers for amplifying the low-level electrical signal to a line-level electrical signal; and

a power supply device connected to the amplifier for supplying operating electrical power for the amplifier and the one or more transducers, wherein the operating power is independent from the line-level electrical signal.

11. The device of claim 10, wherein the one or more transducers comprises one condenser element.

12. The device of claim 10, wherein the one or more transducers comprises two or more condenser elements.

13. The device of claim 12, wherein the two or more condenser elements are separated by a predetermined distance.

14. The device of claim 13, wherein the predetermined distance is approximately 12 inches.

15. The device of claim 10, wherein the amplifier provides two stages of amplification.

16. The device of claim 15, wherein each of the two stages of amplification provides a gain of approximately 15 dB.

17. The device of claim 12, wherein the two or more condenser elements are offset from each other at an angle of approximately 0 degrees to 90 degrees.

18. The device of claim 17, wherein the two or more condenser elements are offset from each other at an angle of approximately 30 degrees.

19. A sound capture device and recording system, comprising:

one or more transducers for converting one or more audio signals to one or more low-level electrical signals and directly inputting the one or more low-level electrical signals to one or more first amplifiers connected to the one or more transducers for amplifying the one or more low-level electrical signals to one or more line-level electrical signals;

one or more power supply devices connected to the one or more first amplifiers for supplying operating electrical power for the one or more first amplifiers and the one or more transducers, wherein the operating power is independent from the one or more line-level electrical signals;

one or more first volume control devices connected to the one or more first amplifiers for receiving the one or more line-level electrical signals and adjusting a volume of one or more line-level electrical signals;

a summing device connected to the one or more first volume control devices for receiving the one or more line-level electrical signals and summing the one or more line-level electrical signals to provide a master line-level electrical signal;

a second volume control connected to the second amplifier for receiving the master line-level

electrical signal and adjusting a volume of the master line-level electrical signal;

a second amplifier connected to the second volume control for receiving the master line-level electrical signal and providing a gain to the master line-level electrical signal;

a recorder connected to the second amplifier for recording the master line-level electrical signal.

20. The system of claim 19, further comprising an analog to digital converter connected between the second amplifier and the recorder, wherein the analog to digital converter converts the master line-level electrical signal from an analog signal to a digital signal.

21. The system of claim 19, wherein the recorder is an analog recorder.

22. A sound capture device, comprising:

a means for converting an audio signal to a low-level electrical signal and inputting the low-level electrical signal to a means for amplifying the low-level electrical signal to a line-level electrical signal; and

a means for supplying power connected to the means for amplifying, wherein the means for supplying

power supplies electrical operating power for the means for amplifying and the means for converting, wherein the electrical operating power is independent from the line-level electrical signal.

23. The device of claim 22, wherein the means for converting comprises one condenser element.

24. The device of claim 22, wherein the means for converting comprises two or more condenser elements.

25. The device of claim 24, wherein the two or more condenser elements are separated by a predetermined distance.

26. The device of claim 25, wherein the predetermined distance is approximately 12 inches.

27. The device of claim 22, wherein the means for amplifying provides two stages of amplification.

28. The device of claim 27, wherein each of the two stages of amplification provides a gain of approximately 15 dB.

29. The device of claim 24, wherein the two or more condenser elements are offset from each other at an angle of approximately 0 degrees to 90 degrees.

30. The device of claim 29, wherein the two or more condenser elements are offset from each other at an angle of approximately 30 degrees.

31. A sound capture device and sound recognition system, comprising:

one or more transducers for converting an audio signal comprising one or more sounds to a low-level electrical signal and directly inputting the low-level electrical signal to an amplifier connected to the one or more transducers for amplifying the low-level electrical signal to a line-level electrical signal;

a power supply device connected to the amplifier for supplying operating electrical power for the amplifier and the one or more transducers, wherein the operating power is independent from the line-level electrical signal;

an analog to digital converter connected to the amplifier for receiving the line-level electrical signal and converting the line-level electrical signal from an analog signal to a digital signal; and

a sound recognition device connected to the analog to digital converter for receiving the digital signal and identifying the one or more sounds.

32. The system of claim 31, wherein the one or more sounds comprises one or more spoken words.

33. The system of claim 31, wherein the one or more sounds comprises one or more musical notes.

34. A method of capturing sound, comprising the steps of:

converting an audio signal to a low-level electrical signal; and

amplifying the low-level electrical signal to provide a line-level electrical signal.

35. The method of claim 34, wherein the step of amplifying the low-level electrical signal to a line-level electrical signal provides two stages of amplification.

36. The method of claim 35, wherein each of the two stages of amplification is approximately 15 dB.

37. A method of recording sound, comprising the steps of:

converting an audio signal to a low-level electrical signal;

amplifying the low-level electrical signal to provide a line-level electrical signal; and

recording the line-level electrical signal.

38. The method of claim 37, wherein the step of amplifying the low-level electrical signal to a line-level electrical comprises two stages of amplifying.

39. The method of claim 38, wherein each of the two stages of amplifying is approximately 15 dB.

ABSTRACT

A sound capture device comprising one or more transducers for converting an audio signal to a low-level electrical signal and directly providing this signal to an amplifier for amplifying the low-level signal to a line-level electrical signal. In one embodiment of the present invention, a power supply is directly connected to the amplifier for supplying operating power that is independent of the audio signal.

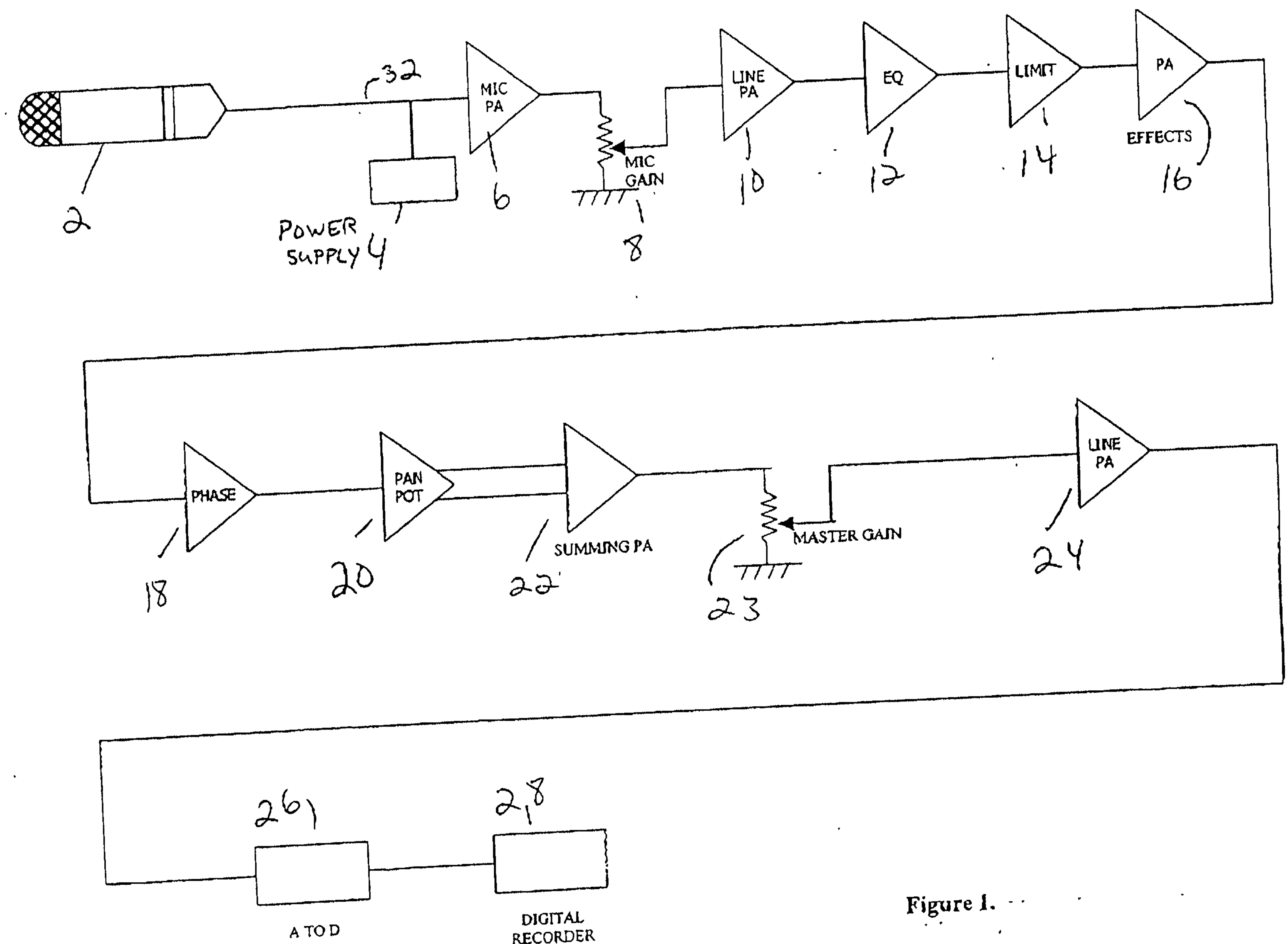


Figure 1.

(Prior Art)

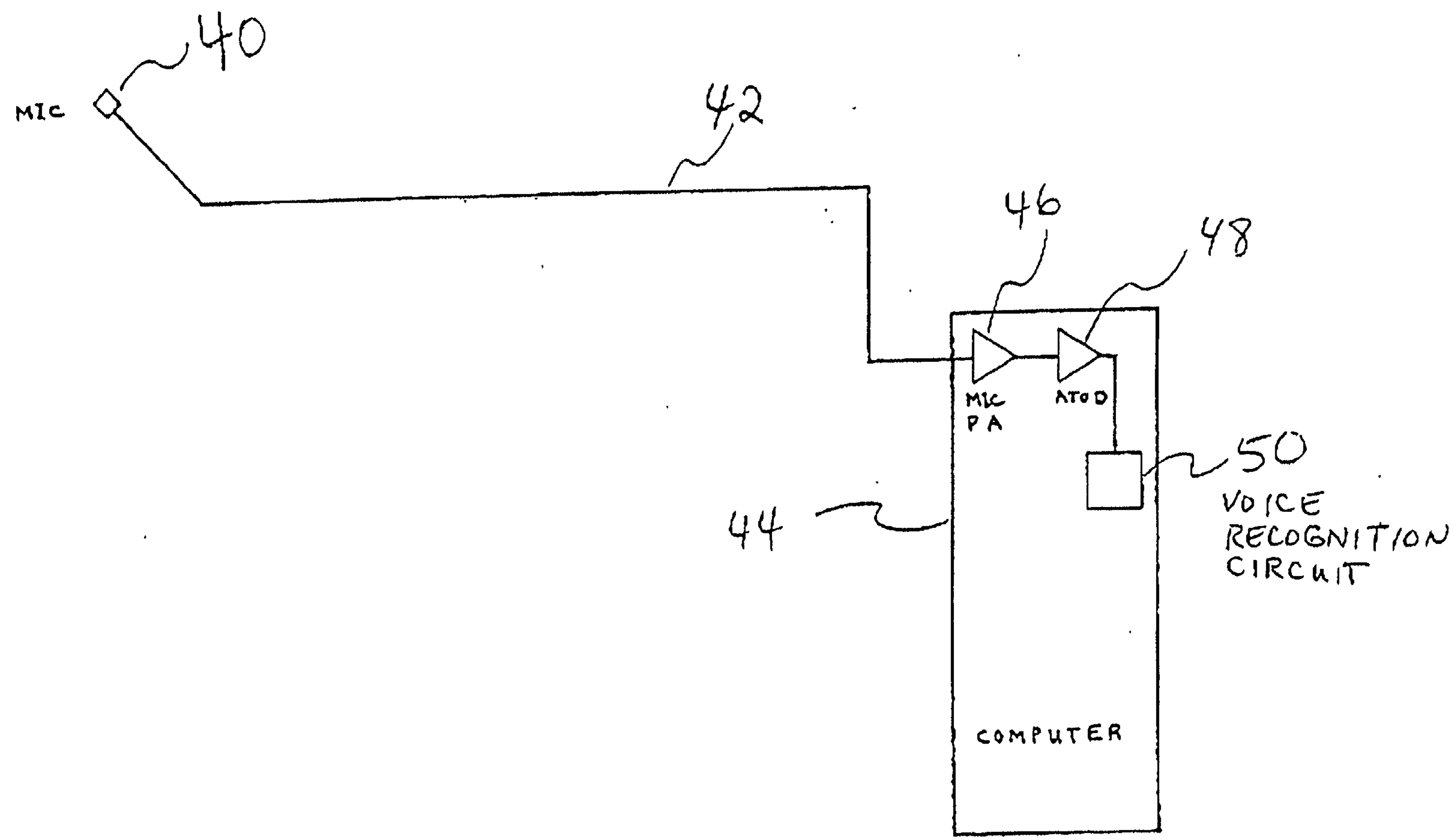


Fig. 2
(Prior Art)

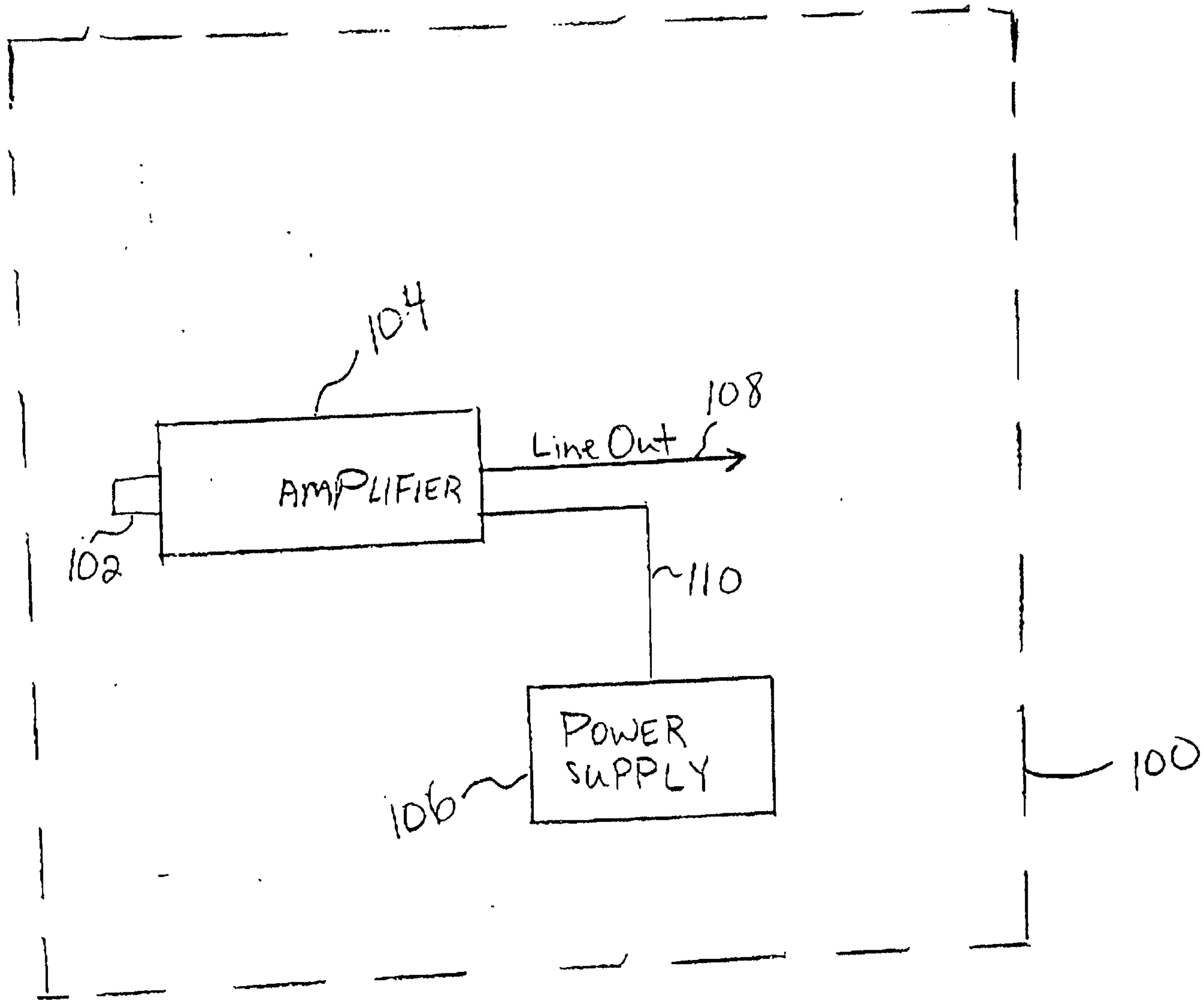


Fig 3

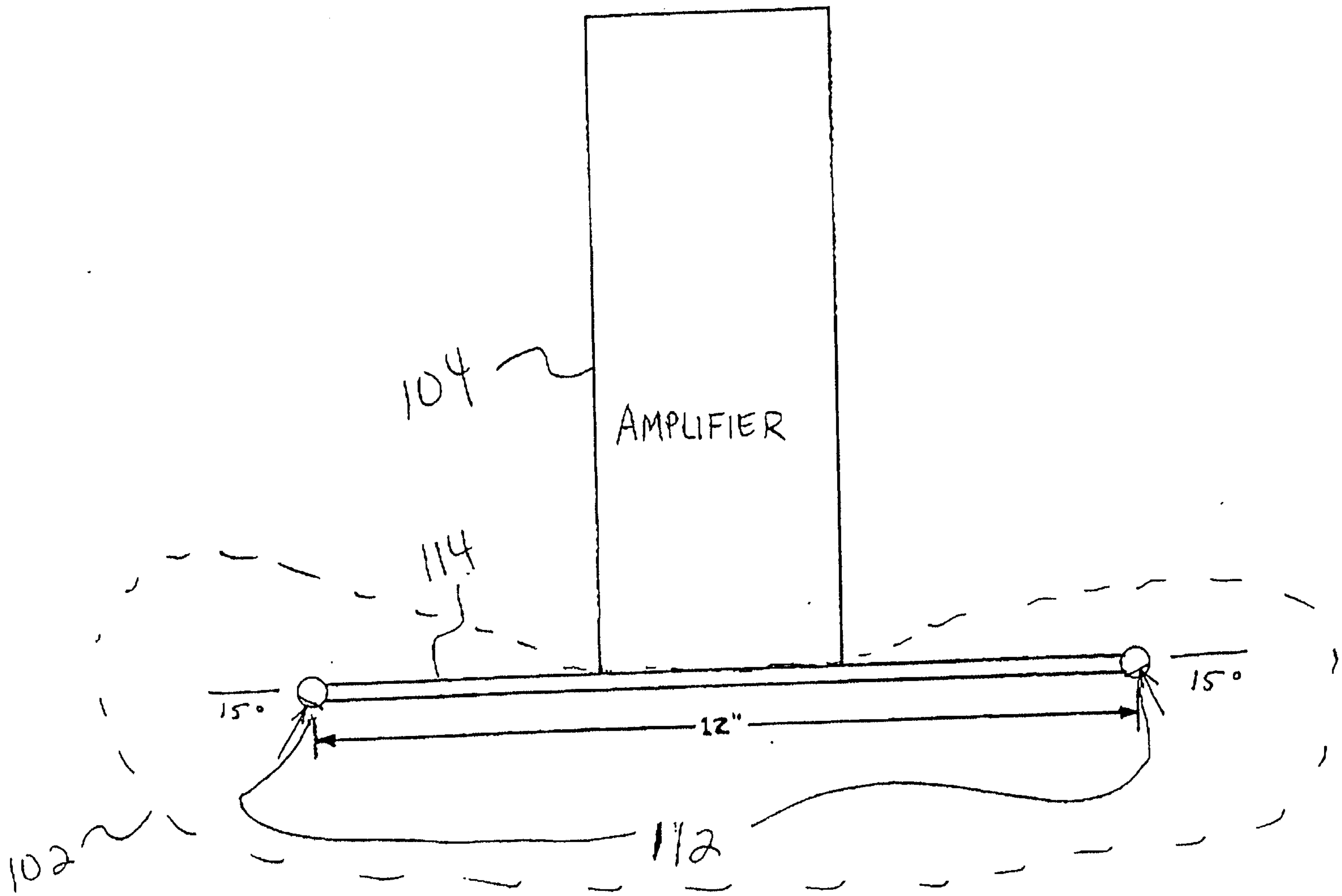
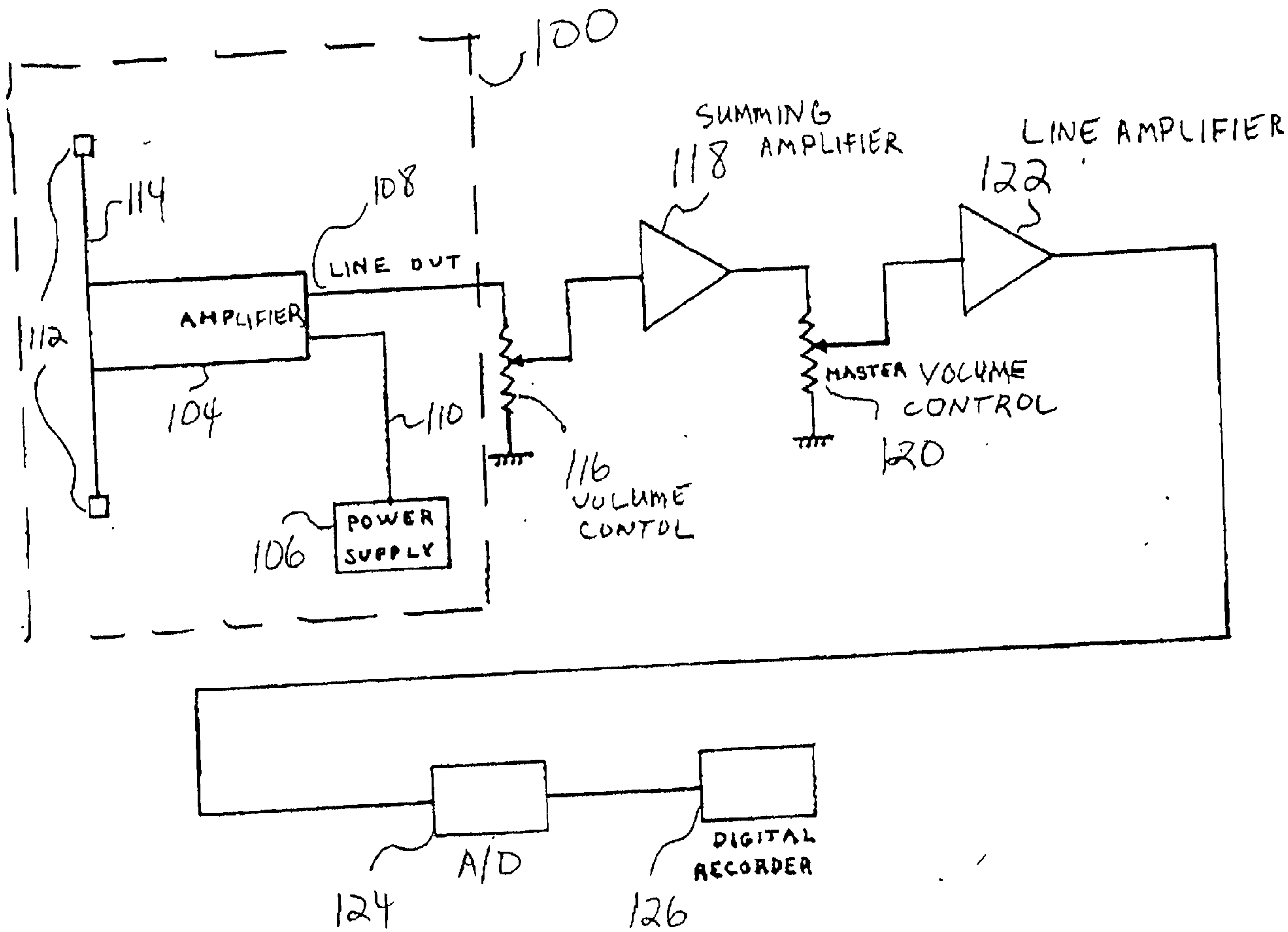


Fig. 4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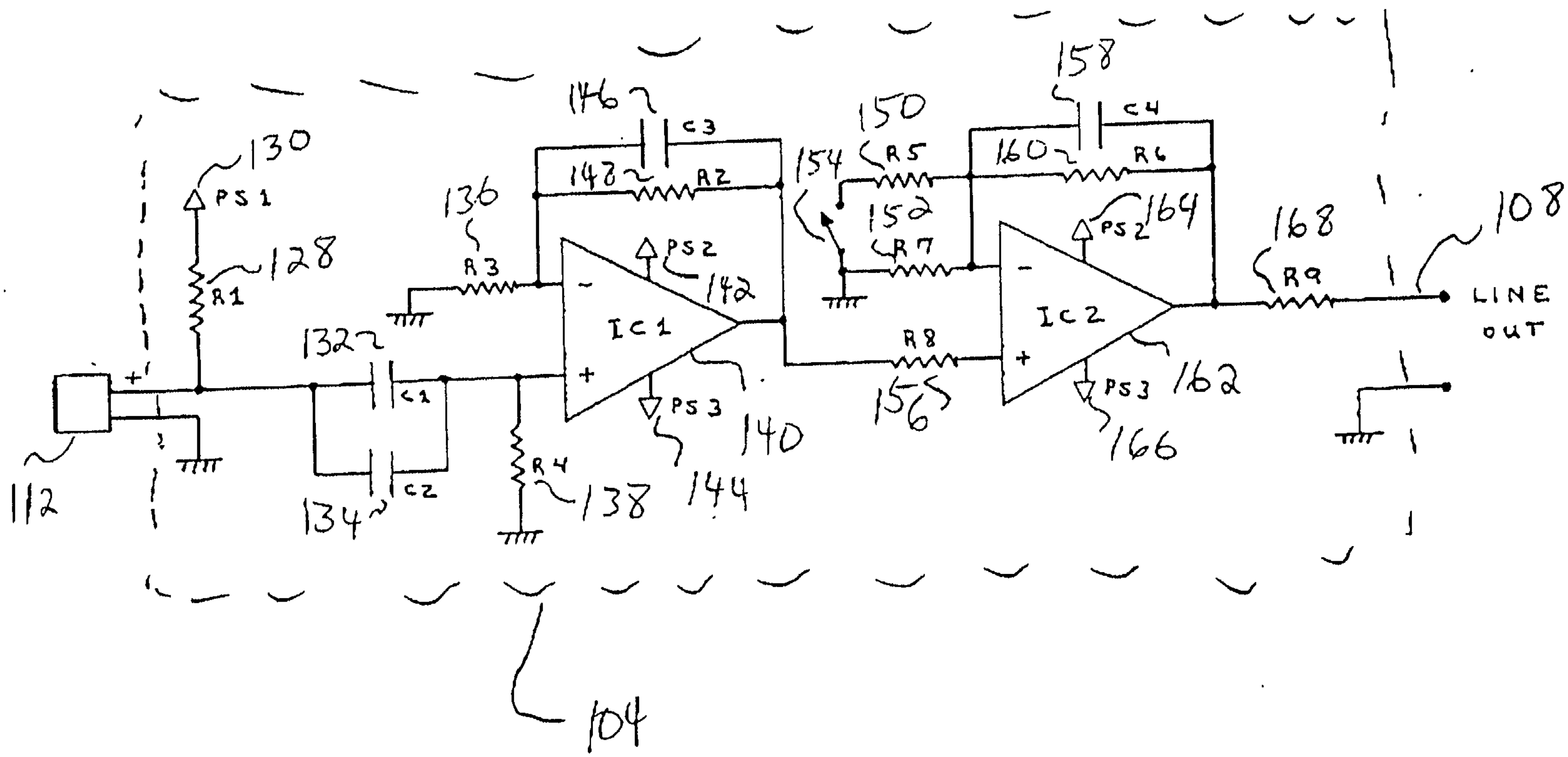
LINE AMPLIFIER
122
114
108
LINE OUT
AMPLIFIER
112
104
110
MASTER VOLUME CONTROL
120
116
VOLUME CONTOL
POWER SUPPLY
106
A/D
124
DIGITAL RECORDER
126

Figure 5



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C1
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C2
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150
R5
154
152
R7
R8
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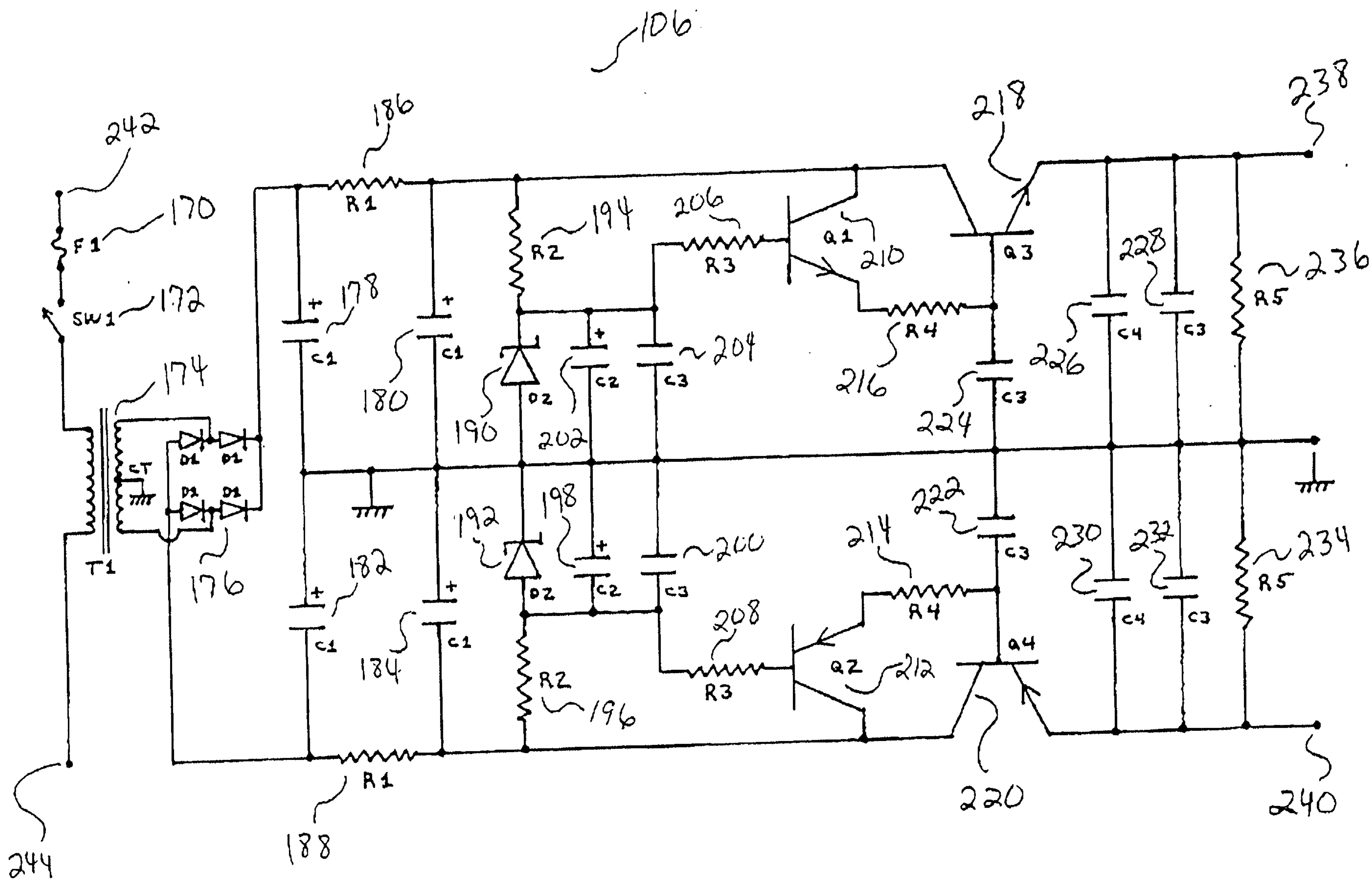
Fig. 6



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+
R2
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DZ
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192
C2
202
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C3
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R3
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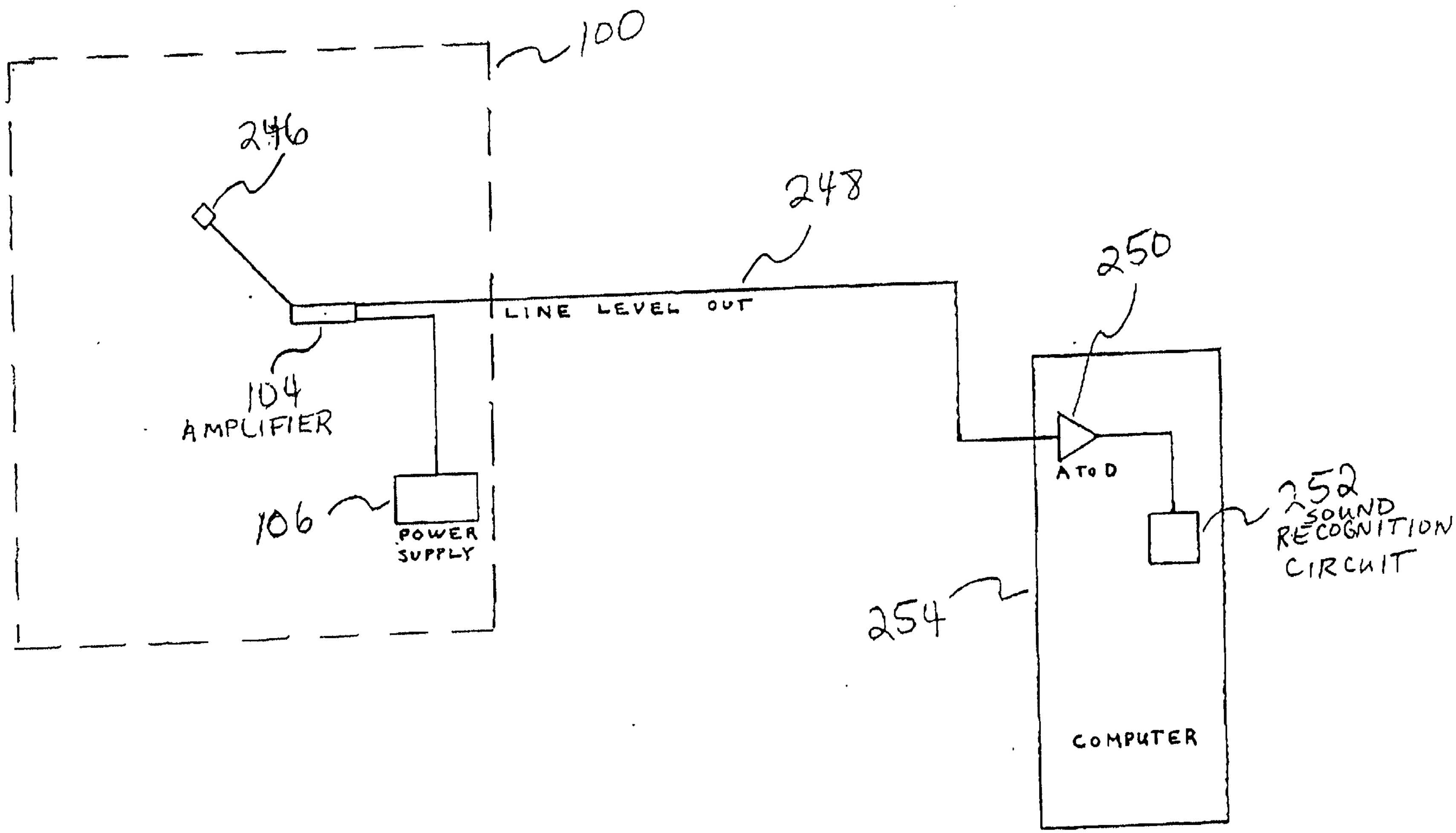
Fig. 7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POWER SUPPLY
252
SOUND RECOGNITION CIRCUIT
254
COMPUTER

Fig. 8

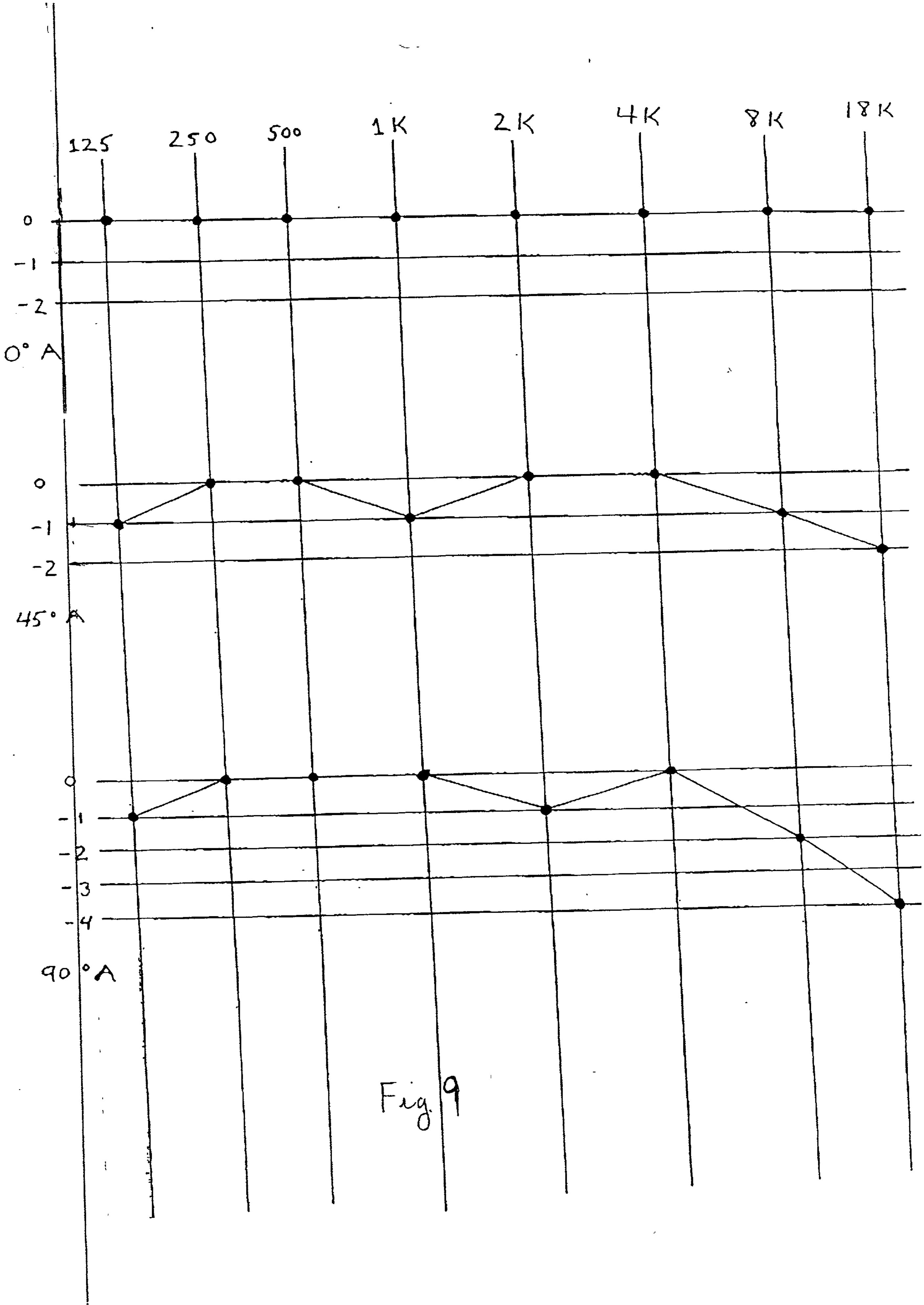

Fig. 9

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☑ *Scanned copy is best available.* There are 33 pages of specification. Not 34